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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/725,294	12/01/2003	Toru Marumoto	9333/360	2919

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ALPINE/BHGL
P.O. Box 10395
Chicago, IL 60610

EXAMINER	
COLUCCI, MICHAEL C	
ART UNIT	**PAPER NUMBER**
2626	
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/725,294	MARUMOTO ET AL.	
	Examiner	Art Unit	
	Michael C. Colucci	2626	

-- *The MAILING DATE of this communication appears on the cover sheet with the correspondence address* --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE _3_ MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) _1-17_ is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) _1-17_ is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on _01 December 2003_ is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☒ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _12/01/2003_.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Continued Examination Under 37 CFR 1.114

1. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on November 28, 2007 has been entered.

Claim Rejections - 35 USC § 102

2. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

3. Claims 1-7, 9, 11-13, and 16 rejected under 35 U.S.C. 102(b) as being anticipated by Urbanski, US 5,544,250 A (hereinafter Urbanski).

Re claims 1 and 4, Urbanski teaches a speech communication apparatus for bi-directional speech communications, comprising:

a speaker (col 1 lines 42-56);

a microphone (Fig. 1 item 101);

background sound level measurement means for extracting background sound from an output signal of the microphone and for measuring a level of the extracted background sound from the output signal (col 1 lines 21-29);

received-speech clarifying means for adjusting a gain (Fig. 2 item 207) for a received-speech signal to be output by the speaker based on the level of the background sound from the output signal measured by the background sound level measurement means (col 1 lines 21-29);

wherein the speech communication apparatus does not comprise more than one microphone (Fig. 1 item 101).

NOTE: Fig. 1 illustrates a noise suppression system depicting the use of only one microphone 101.

NOTE: For purposes of prior art, the proximity effect is construed to be both functionally equivalent and equally effective as noise produced from low frequency components picked up by the microphone in a changing environment, where the response of a system would be handled by an adaptive frequency response such as the adjustment of gain and reduction of noise.

Re claims 2, 6, 12, and 16, Urbanski teaches the speech communication apparatus of claim 1, further comprising:

received-speech-level measurement means for measuring a level of the received-speech signal at each predetermined frequency band (col 1 lines 21-41 & Fig. 3),

wherein the background sound level measurement means measures the level of the background sound in each predetermined frequency band (col 1 lines 21-41 & Fig. 3) and the received-speech clarifying means performs loudness compensation in which the gain for the received-speech signal is adjusted (Fig. 2 item 207) in each predetermined frequency band (col 1 lines 21-41 & Fig. 3) such that received speech output by the speaker is heard at almost the same intensity in the human auditory sense irrespective of the level of the background sound (col 1 lines 21-41 & Fig. 3), and the resultant signal is output to the speaker as the received speech (col 1 lines 42-56).

Re claims 3, 7, 13, and 17, Urbanski teaches the speech communication apparatus of claim 1, wherein the speech communication apparatus is a portable, mobile telephone for performing the speech communications by radio communication (col 1 lines 42-56).

Re claim 5 and 11, Urbanski teaches the speech communication apparatus of claim 4, wherein the microphone is a unidirectional or bi-directional microphone (Fig. 1 item 101).

NOTE: The term microphone disclosed in general, is encompassing of any microphone polar pattern including but not limited to unidirectional or bidirectional microphone. Therefore, it would be necessary to utilize a microphone with a particular polar pattern as a matter of choice if and when directional preference is desirable. Urbanski teaches a general microphone such as a unidirectional or bidirectional

microphone on a mobile cellular phone (col 1 lines 42-56), where a bidirectional microphone could consist of two unidirectional microphones facing opposite one another.

Re claim 9, Urbanski teaches the speech communication apparatus of claim 8, further comprising:

transmission means for transmitting an output of the transmission-speech filter as a transmission-speech signal from the speech communications apparatus (col 1 lines 13-20).

Claim Rejections - 35 USC § 103

4. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

5. Claims 8, 10, 14, and 15 rejected under 35 U.S.C. 103(a) as being unpatentable over Urbanski, US 5,544,250 A (hereinafter Urbanski) in view of Todter et al US 5937070 A (hereinafter Todter).

Re claims 14 and 8, Urbanski teaches a speech communication apparatus for bi-directional speech communications, comprising:

a speaker for outputting received speech (col 1 lines 42-56)

a microphone for collecting speech to be transmitted (Fig. 1 item 101);

background sound level measurement calculator operable to measure a level of background sound (col 1 lines 21-29);

a received-speech clarifying section operable to adjust a gain for the received speech to be outputted by the speaker based on the level of the background sound measured by the background sound level measurement calculator (col 1 lines 29-41), wherein the received speech to be outputted by the speaker is not received by a microphone of the speech communication apparatus (Fig. 1 item 101).

NOTE: For purposes of prior art, the microphone not collecting the output from the speaker is construed to be both functionally equivalent and equally effective as a speech communication system that does not implement feedback from a speaker to a microphone (Fig. 1 item 1). Urbanski does not teach any methods of feedback from final output to input microphone. Additionally, for purposes of prior art, the proximity effect is construed to be both functionally equivalent and equally effective as noise produced from low frequency components picked up by the microphone in a changing environment, where the response of a system would be handled by an adaptive frequency response such as the adjustment of gain and reduction of noise.

However, Urbanski fails to teach a delay section operable to delay an output of a first background-sound microphone by a period of time corresponding to a delay time between transmission speech mixed into the output of the first background-sound

microphone and transmission speech mixed into an output of a second background-sound microphone (Todter col 15 lines 1-15 & Fig. 12 item 72),

an adaptive filter (Todter col 13 lines 31-49) operable to estimate transmission of speech mixed into the output of the delay section (Todter col 15 lines 1-15 & Fig. 12 item 72),

an adder operable to subtract the transmission speech estimated by the adaptive filter from an output of the delay section (Todter col 15 lines 1-15 & Fig. 12 items 72 and 73),

a background sound level calculation section operable to calculate a level of an output of the adder and for outputting the result as the level of the background sound (Todter col 15 lines 1-15 & Fig. 12 items 74 and summation node prior to output just below item 74).

Todter teaches a high pass frequency segment that provides for a weighted and phase-shifted sum of the "n" microphone signals; containing both phase correlated speaker's voice and uncorrelated external noise signals. The invention allows for the adjustment of signal weighting and phase shifting to amplify the speakers voice signal and attenuate the external noise. (59) The high frequency sections comprises a high pass filter 80 and a plurality of gain and delay blocks 81, 82, respectively, as well as a plurality of positive summing circuits 83. The low pass frequency filter may also comprise a plurality of circuits 71, 72, 73. (60) The low pass frequency segment of the invention provides for a weighted and phase-shifted subtraction of noise from the mouthpiece microphone signal. The invention allows for adjustment of gain weighting

and phase shift to find the optimum improvement in signal to noise ratio, in any specific reverberant noise environment. (61) The block 60 further provides for the weighted summing via summing circuit 90 of low passed and high passed signals to reconstitute the total enhanced signal.

Additionally, Todter teaches an adaptive control block 9 that compares the noise cancellation signal output from the summing circuit 13 with the microphone signal to detect residual noise, and controls the gain compensation 20 and phase compensation 21 to keep the residual noise to a minimum. Also by negatively summing the audio signal into the adaptive control loop the loop will affect the compensation of the cancellation loop block to provide a high fidelity output signal, notwithstanding quality of components, as discussed in the preamble.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention a delay section utilizing several microphones where delayed speech from several microphones are mixed together, using an adaptive filter in for the subtraction of speech from a delay section. Delaying segments of speech would be necessary to add high pass filtered and low pass filtered components, where low pass components can be weighted and summed in addition to high pass components, to reduce the amount of noise in a signal. In a phase shifting environment, summation of delayed components from several microphones would allow for reduced noise, adjustable gain, and optimal signal to noise ratio conditions in a noisy or non-noisy environment.

Re claim 10, Urbanski teaches a speech communication apparatus for bi-directional speech communications, provided with a handset having at a front face a speaker for outputting received speech and a transmission-speech microphone for collecting speech to be transmitted (col 1 lines 42-56), the speech communication apparatus comprising:

background sound level measurement means for measuring a level of an output from the background-sound microphone as a background-sound level (col 1 lines 21-29);

received-speech clarifying means for adjusting a gain for received speech (Fig. 2 item 207) that is output from the speaker based on the background-sound level measured by the background sound level measurement means, wherein the received speech that is output from the speaker is not received at a microphone of the speech communication apparatus (col 1 lines 29-41).

NOTE: For purposes of prior art, the microphone not collecting the output from the speaker is construed to be both functionally equivalent and equally effective as a speech communication system that does not implement feedback from a speaker to a microphone (Fig. 1 item 1). Urbanski does not teach any methods of feedback from final output to input microphone.

However, Urbanski fails to teach a background-sound microphone disposed at the rear face of the handset at almost the same height as the speaker, for collecting background sound (Todter col 8 lines 50-65 & col 9 lines 12-25);

Todter teaches a plurality of pick ups or microphones are provided (e.g. in a telephone handset) at known positions in relation to the source of the desired audio signal, so that all microphones will receive the desired audio signal (albeit possibly with gain and phase differences) and all microphones will receive the extraneous noise signal (again possibly with gain and phase differences). The signals from each microphone may then be separately processed to provide electrical noise cancellation and added (possibly with appropriate weighting) to give the noise cancelled audio signal. The noise cancelling processing may make use of known propagation characteristic differences between the ambient noise field and the desired audio signal acoustic field.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention a background noise microphone at the rear of a handset at almost the same height of the speaker, to collect background noise. Using a microphone and speaker on a handset would be necessary for a user of the headset to be able to communicate and transmit data in response to incoming data. Using a microphone one or more microphones a headset would allow for the acquisition of the same noise signal but with various gains and phase differences, where several noise cancellation operations can be performed and summed to produce a desirable speech signal.

Re claim 15, Urbanski fails to teach the speech communication apparatus of [[to]] claim 14, wherein the adaptive filter (Todter col 13 lines 31-49) estimates the transmission speech based on a difference between the output of the delay section and

the transmission speech estimated by the adaptive filter (Todter col 15 lines 1-15 & Fig. 12 items 72 and 73).

Todter teaches a high pass frequency segment that provides for a weighted and phase-shifted sum of the "n" microphone signals; containing both phase correlated speaker's voice and uncorrelated external noise signals. The invention allows for the adjustment of signal weighting and phase shifting to amplify the speakers voice signal and attenuate the external noise. (59) The high frequency sections comprises a high pass filter 80 and a plurality of gain and delay blocks 81, 82, respectively, as well as a plurality of positive summing circuits 83. The low pass frequency filter may also comprise a plurality of circuits 71, 72, 73. (60) The low pass frequency segment of the invention provides for a weighted and phase-shifted subtraction of noise from the mouthpiece microphone signal. The invention allows for adjustment of gain weighting and phase shift to find the optimum improvement in signal to noise ratio, in any specific reverberant noise environment. (61) The block 60 further provides for the weighted summing via summing circuit 90 of low passed and high passed signals to reconstitute the total enhanced signal.

Additionally, Todter teaches an adaptive control block 9 that compares the noise cancellation signal output from the summing circuit 13 with the microphone signal to detect residual noise, and controls the gain compensation 20 and phase compensation 21 to keep the residual noise to a minimum. Also by negatively summing the audio signal into the adaptive control loop the loop will affect the compensation of the

cancellation loop block to provide a high fidelity output signal, notwithstanding quality of components, as discussed in the preamble.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention a delay section utilizing several microphones where delayed speech from several microphones are mixed together, using an adaptive filter in for the subtraction of speech from a delay section. Delaying segments of speech would be necessary to add high pass filtered and low pass filtered components, where low pass components can be weighted and summed in addition to high pass components, to reduce the amount of noise in a signal. In a phase shifting environment, summation of delayed components from several microphones would allow for reduced noise, adjustable gain, and optimal signal to noise ratio conditions in a noisy or non-noisy environment.

Conclusion

6. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. US 4420655 A, US 4630305 A, US 5732390 A, US 5937070 A, US 6148078 A, US 6466832 B1, US 20030061050 A1, US 6591234 B1, US 20060120537 A1, US 20070038436 A1.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael C. Colucci whose telephone number is (571)-

270-1847. The examiner can normally be reached on 9:30 am - 6:00 pm, Monday-Friday.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (571)-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

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